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Takizawa et al.

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(54) **BROADCAST STATION SYNCHRONIZATION METHOD AND MOBILE TERMINAL**

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(30) **Foreign Application Priority Data**

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(51) **Int. Cl.**
H04B 7/00 (2006.01)

(52) **U.S. Cl.** **455/502**; 455/503; 370/509; 370/110; 381/107; 381/119; 700/94

(58) **Field of Classification Search** 455/502; 370/509, 110; 381/107, 119; 700/94
See application file for complete search history.

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(57) **ABSTRACT**

A broadcast station synchronization method in a mobile terminal operating with a system clock different from the reference clock of a broadcast station, wherein a clock synchronized with the broadcast station is reproduced according to an internal reference clock corrected by the reference time information from the broadcast station, a difference between an elapsed time at a predetermined time interval and the audio reproduction time is detected, the audio reproduction time is adjusted to cancel the difference, and the PCM transmission for reproducing the audio uses a signal indicating the PCM data transmission completion as a trigger for latching the internal reference clock count unit reproducing the clock synchronized with the broadcast station so as to acquire the clock time synchronized with the broadcast station of the same period as the audio reproduction time per a predetermined period, thereby performing synchronization with the broadcast station.

3 Claims, 8 Drawing Sheets

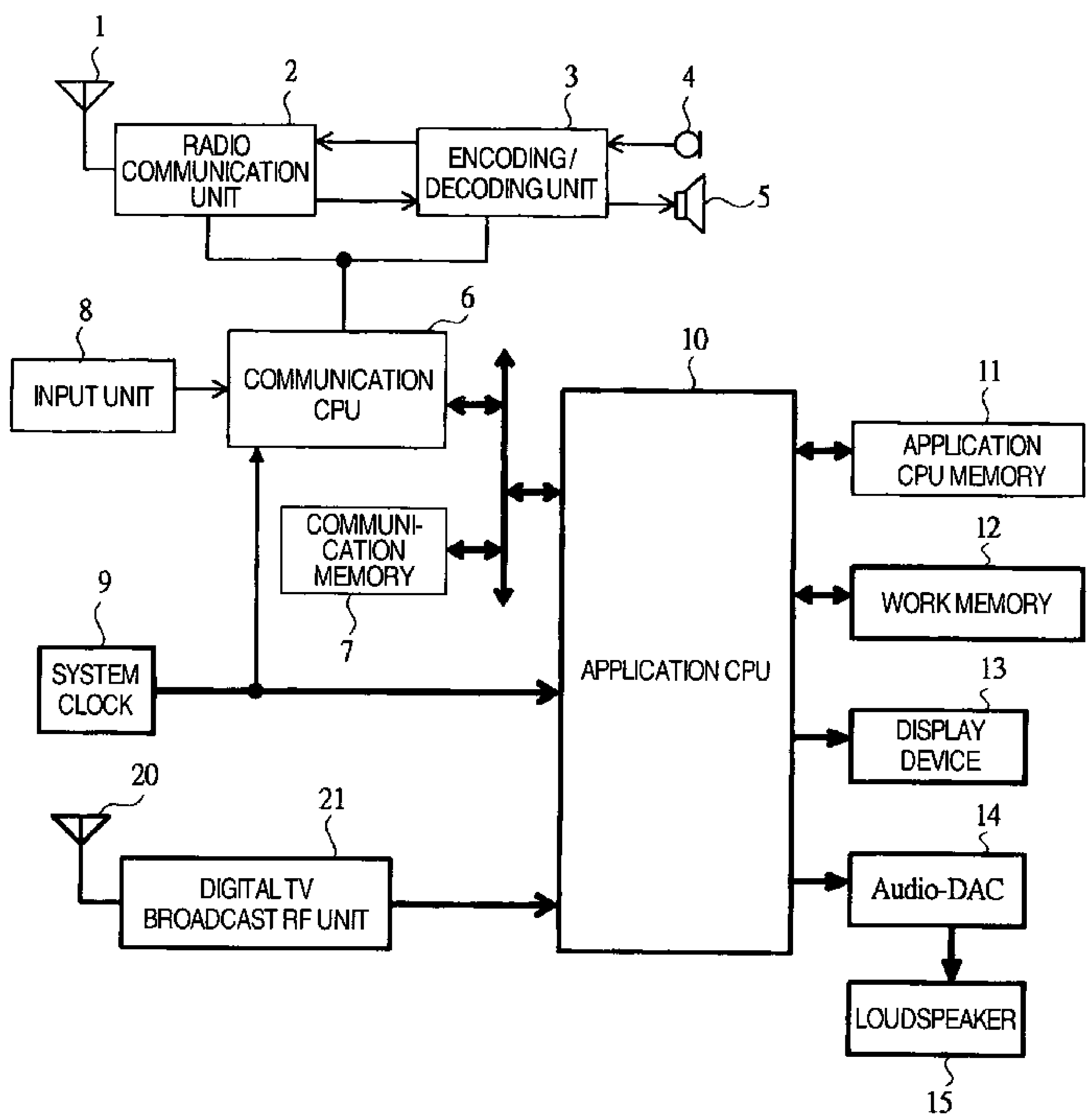


FIG.1

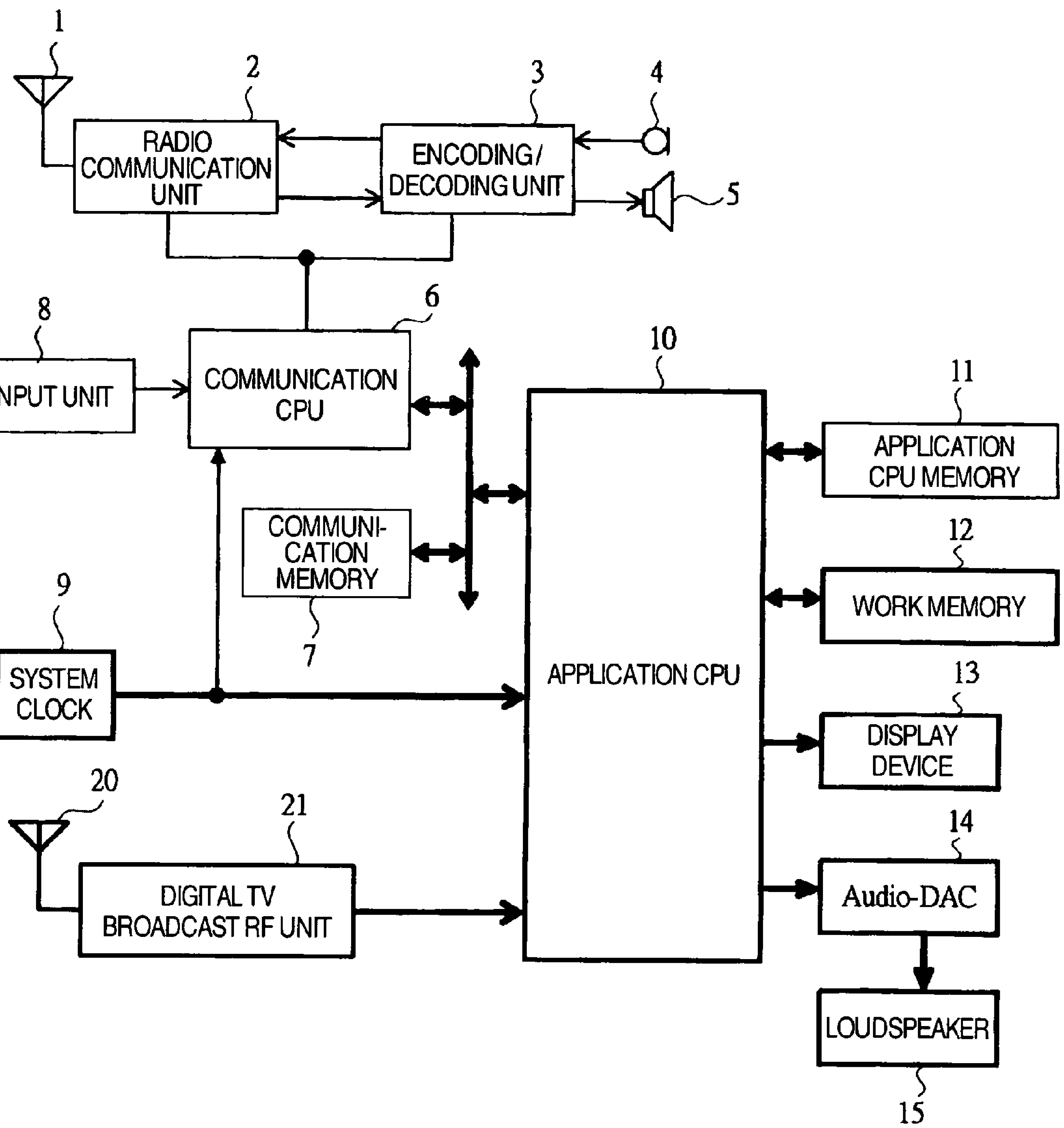
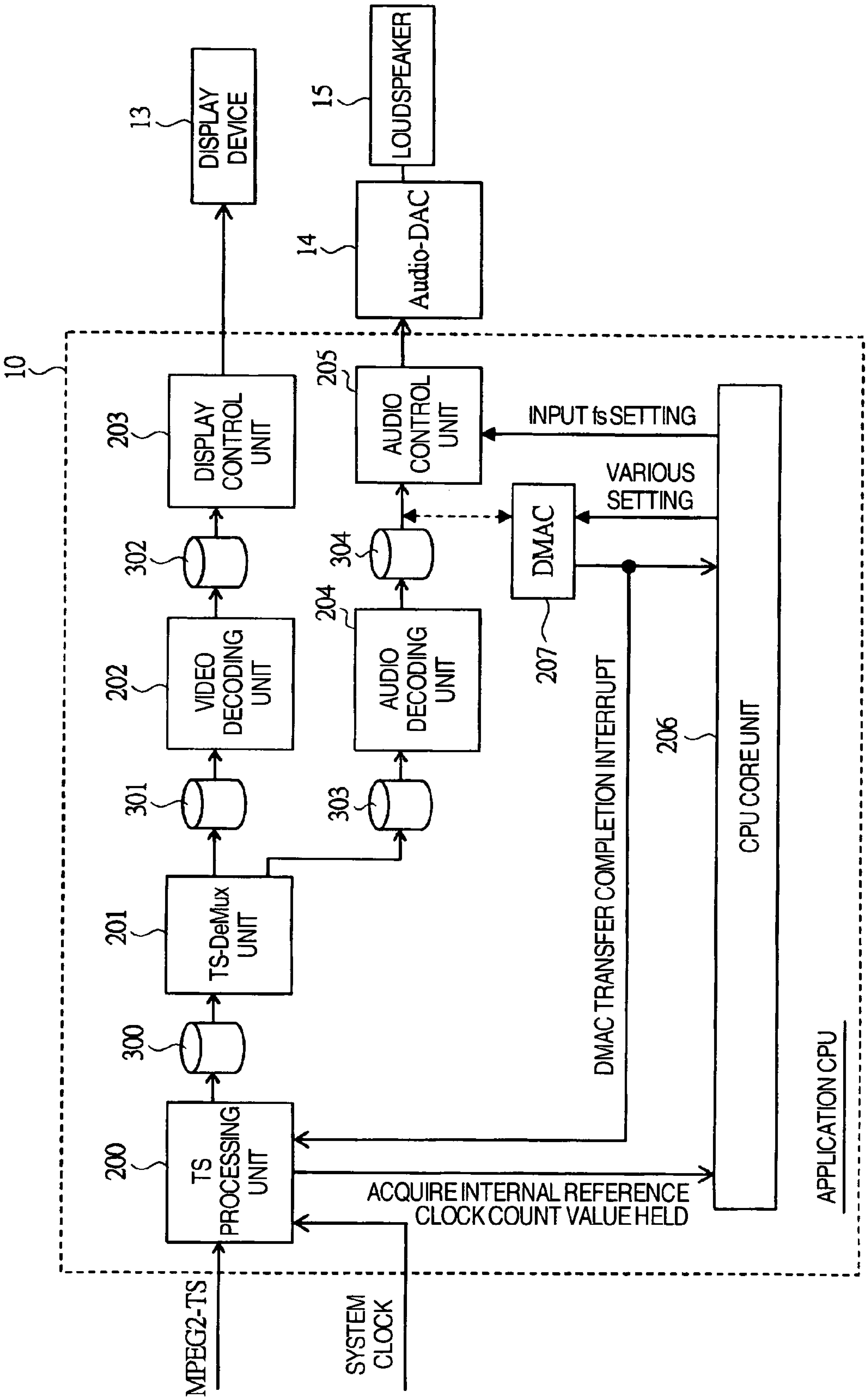

1
2
RADIO COMMUNICATION UNIT
3
ENCODING/ DECODING UNIT
4
5
8
INPUT UNIT
6
COMMUNICATION CPU
10
APPLICATION CPU
11
APPLICATION CPU MEMORY
COMMUNI-CATION MEMORY
7
12
WORK MEMORY
9
SYSTEM CLOCK
13
DISPLAY DEVICE
20
21
DIGITAL TV BROADCAST RF UNIT
14
Audio-DAC
LOUDSPEAKER
15

FIG.2



10
MPEG2-TS
SYSTEM CLOCK
200
TS PROCESSING UNIT
300
201
TS-DeMux UNIT
301
303
202
VIDEO DECODING UNIT
302
203
DISPLAY CONTROL UNIT
13
DISPLAY DEVICE
204
AUDIO DECODING UNIT
304
205
AUDIO CONTROL UNIT
14
Audio-DAC
15
LOUDSPEAKER
207
DMAC
VARIOUS SETTING
INPUT fs SETTING
DMAC TRANSFER COMPLETION INTERRUPT
206
CPU CORE UNIT
ACQUIRE INTERNAL REFERENCE CLOCK COUNT VALUE HELD
APPLICATION CPU

FIG.3

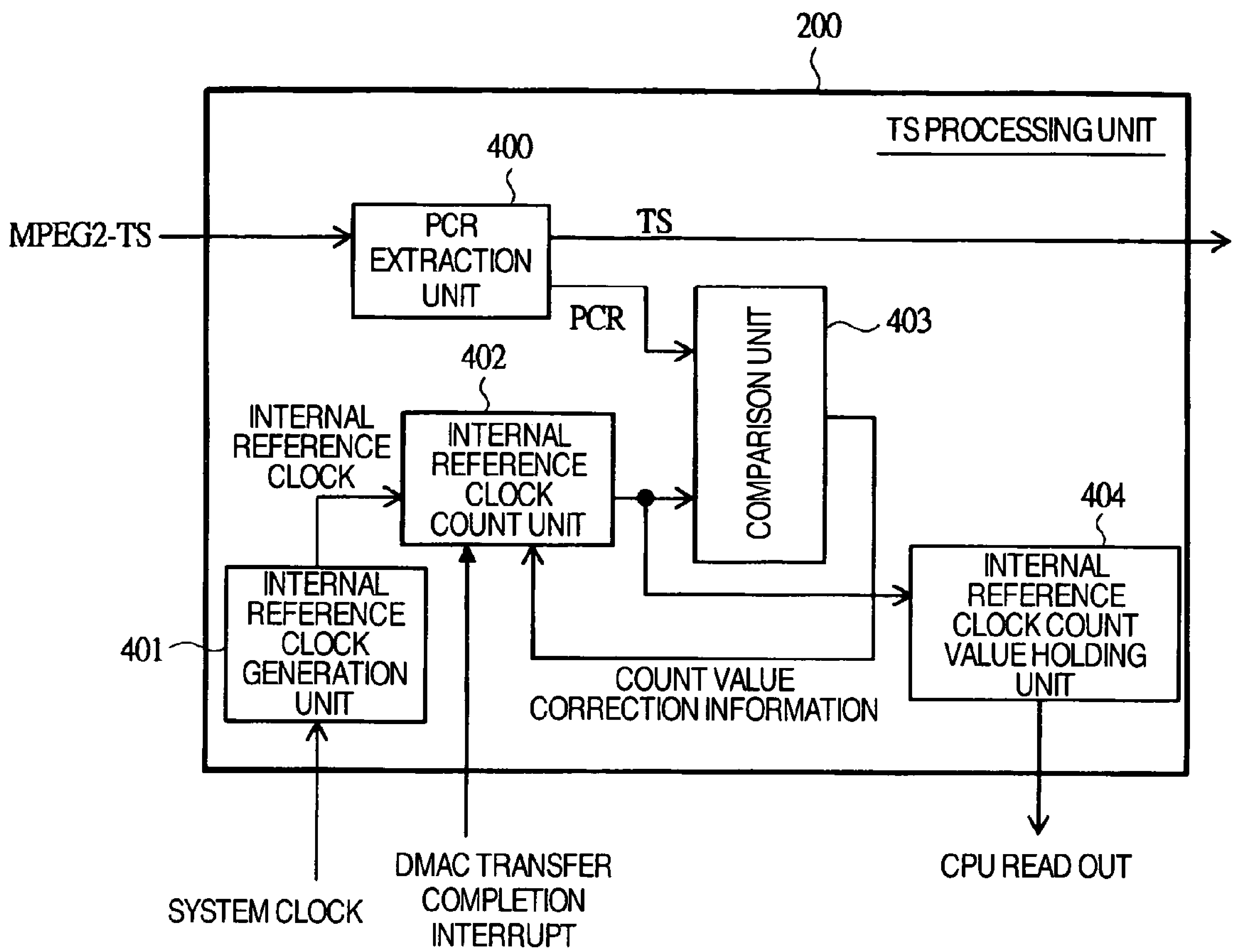

200
TS PROCESSING UNIT
400
MPEG2-TS
PCR EXTRACTION UNIT
TS
PCR
403
COMPARISON UNIT
402
INTERNAL REFERENCE CLOCK
INTERNAL REFERENCE CLOCK COUNT UNIT
404
INTERNAL REFERENCE CLOCK COUNT VALUE HOLDING UNIT
401
INTERNAL REFERENCE CLOCK GENERATION UNIT
COUNT VALUE CORRECTION INFORMATION
SYSTEM CLOCK
DMAC TRANSFER COMPLETION INTERRUPT
CPU READ OUT

FIG.4

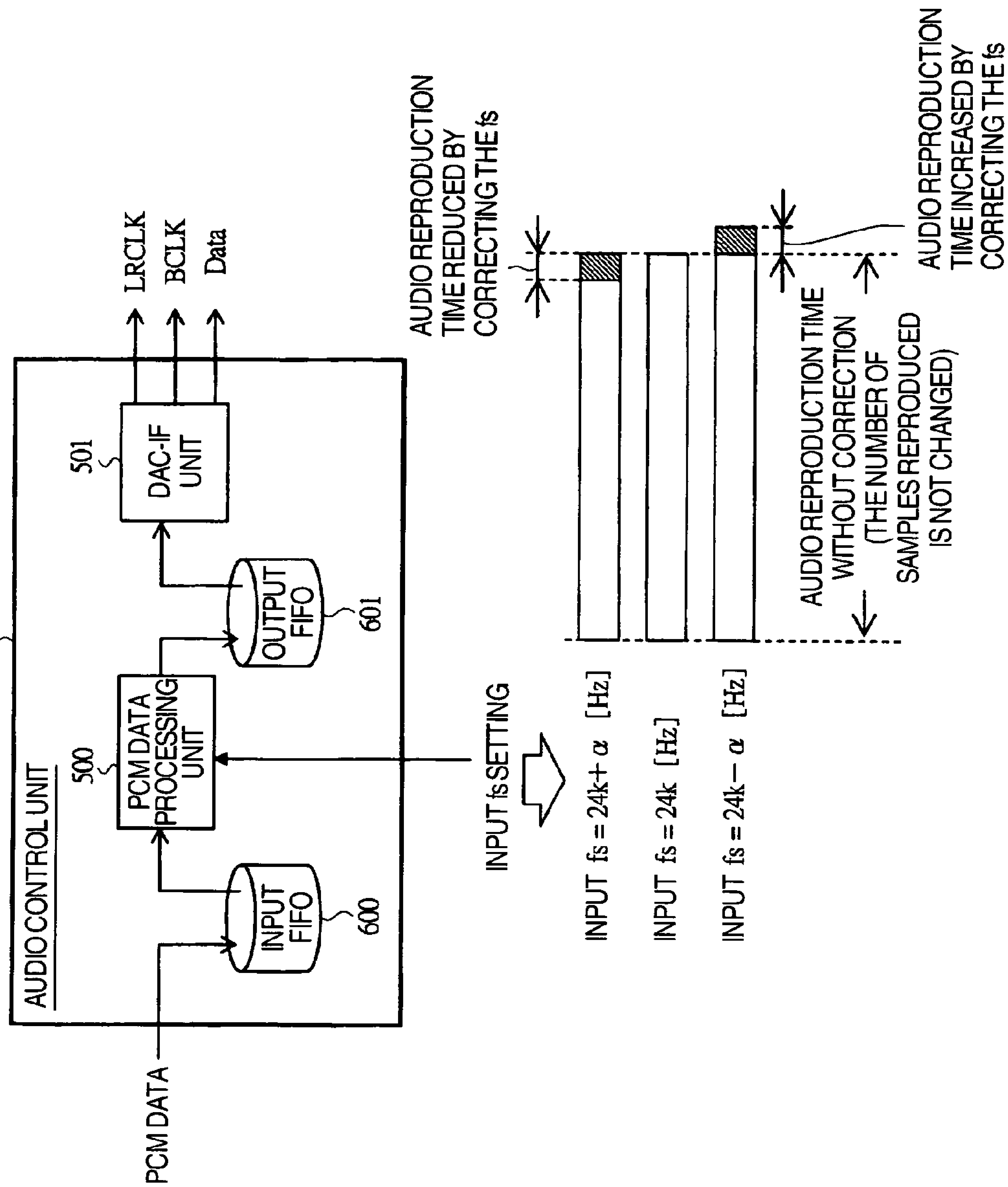

205
AUDIO CONTROL UNIT
PCM DATA
INPUT FIFO
600
500
PCM DATA PROCESSING UNIT
OUTPUT FIFO
601
501
DAC-IF UNIT
LRCLK
BCLK
Data
INPUT fs SETTING
INPUT fs = 24k+α [Hz]
INPUT fs = 24k [Hz]
INPUT fs = 24k−α [Hz]
AUDIO REPRODUCTION TIME REDUCED BY CORRECTING THE fs
AUDIO REPRODUCTION TIME WITHOUT CORRECTION (THE NUMBER OF SAMPLES REPRODUCED IS NOT CHANGED)
AUDIO REPRODUCTION TIME INCREASED BY CORRECTING THE fs

FIG.5

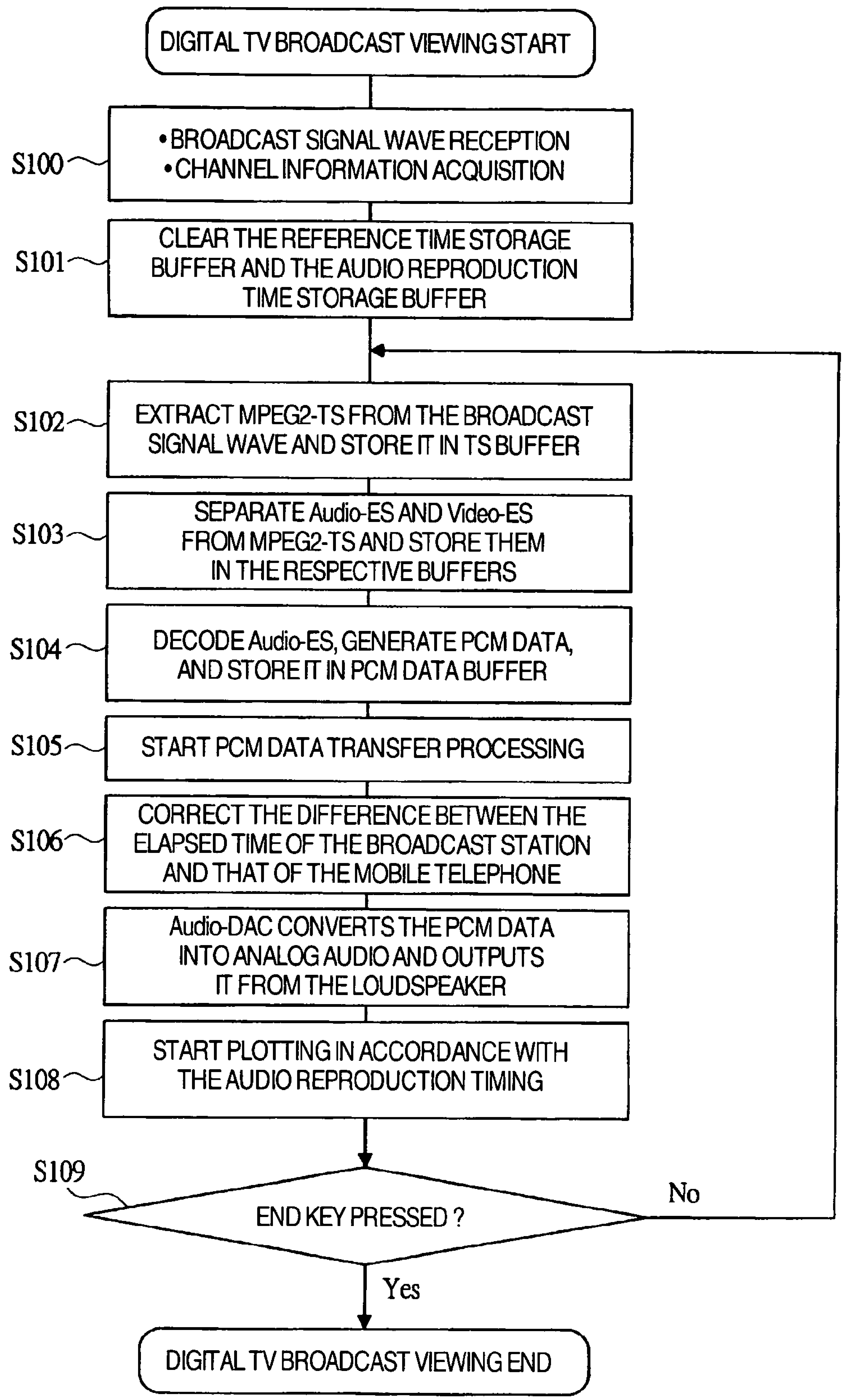
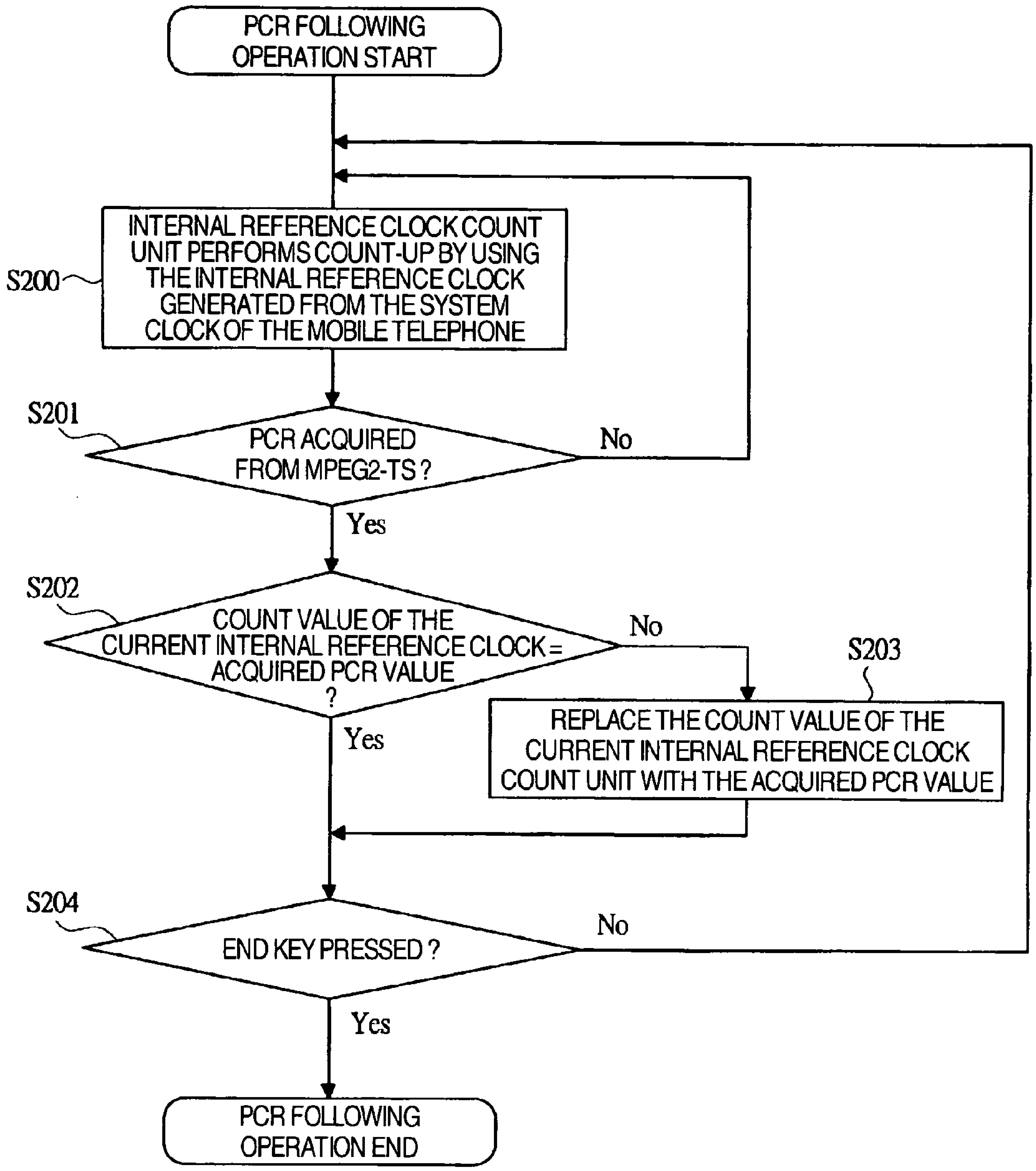

DIGITAL TV BROADCAST VIEWING START
S100
• BROADCAST SIGNAL WAVE RECEPTION
• CHANNEL INFORMATION ACQUISITION
S101
CLEAR THE REFERENCE TIME STORAGE BUFFER AND THE AUDIO REPRODUCTION TIME STORAGE BUFFER
S102
EXTRACT MPEG2-TS FROM THE BROADCAST SIGNAL WAVE AND STORE IT IN TS BUFFER
S103
SEPARATE Audio-ES AND Video-ES FROM MPEG2-TS AND STORE THEM IN THE RESPECTIVE BUFFERS
S104
DECODE Audio-ES, GENERATE PCM DATA, AND STORE IT IN PCM DATA BUFFER
S105
START PCM DATA TRANSFER PROCESSING
S106
CORRECT THE DIFFERENCE BETWEEN THE ELAPSED TIME OF THE BROADCAST STATION AND THAT OF THE MOBILE TELEPHONE
S107
Audio-DAC CONVERTS THE PCM DATA INTO ANALOG AUDIO AND OUTPUTS IT FROM THE LOUDSPEAKER
S108
START PLOTTING IN ACCORDANCE WITH THE AUDIO REPRODUCTION TIMING
S109
END KEY PRESSED ?
No
Yes
DIGITAL TV BROADCAST VIEWING END

FIG.6



PCR FOLLOWING OPERATION START
S200
INTERNAL REFERENCE CLOCK COUNT UNIT PERFORMS COUNT-UP BY USING THE INTERNAL REFERENCE CLOCK GENERATED FROM THE SYSTEM CLOCK OF THE MOBILE TELEPHONE
S201
PCR ACQUIRED FROM MPEG2-TS ?
No
Yes
S202
COUNT VALUE OF THE CURRENT INTERNAL REFERENCE CLOCK = ACQUIRED PCR VALUE ?
No
S203
REPLACE THE COUNT VALUE OF THE CURRENT INTERNAL REFERENCE CLOCK COUNT UNIT WITH THE ACQUIRED PCR VALUE
Yes
S204
END KEY PRESSED ?
No
Yes
PCR FOLLOWING OPERATION END

FIG.7

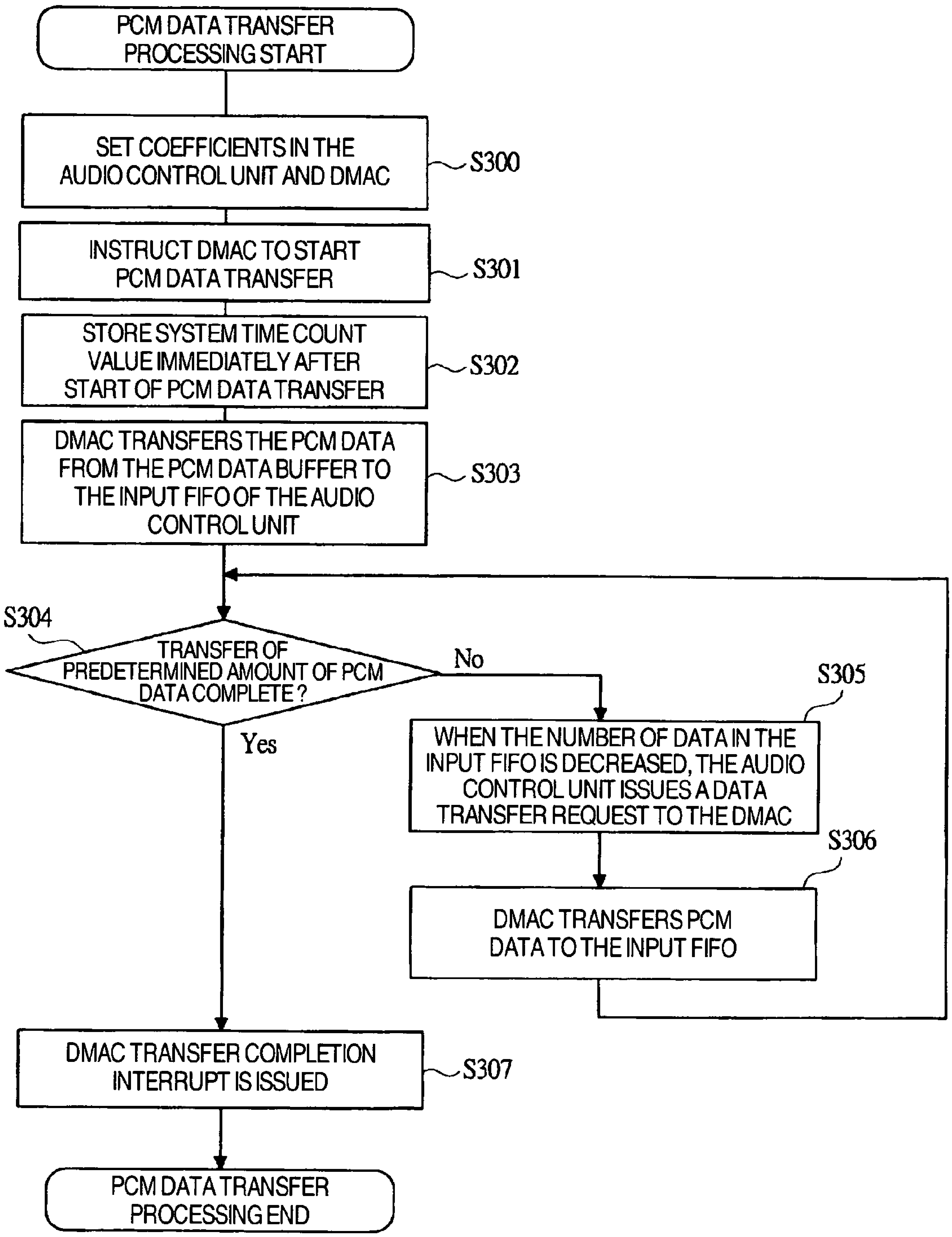
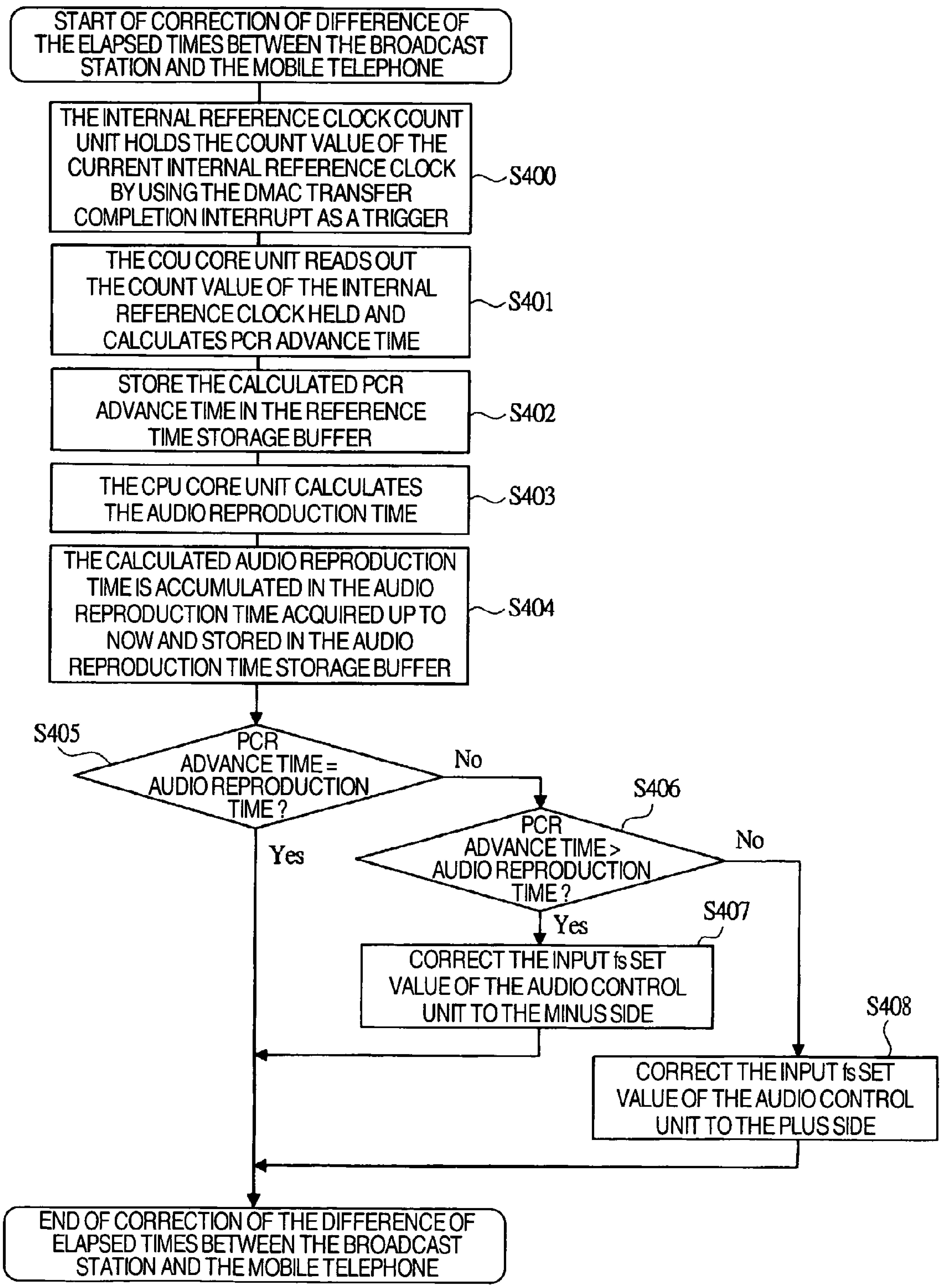

PCM DATA TRANSFER PROCESSING START
SET COEFFICIENTS IN THE AUDIO CONTROL UNIT AND DMAC
S300
INSTRUCT DMAC TO START PCM DATA TRANSFER
S301
STORE SYSTEM TIME COUNT VALUE IMMEDIATELY AFTER START OF PCM DATA TRANSFER
S302
DMAC TRANSFERS THE PCM DATA FROM THE PCM DATA BUFFER TO THE INPUT FIFO OF THE AUDIO CONTROL UNIT
S303
S304
TRANSFER OF PREDETERMINED AMOUNT OF PCM DATA COMPLETE ?
No
S305
WHEN THE NUMBER OF DATA IN THE INPUT FIFO IS DECREASED, THE AUDIO CONTROL UNIT ISSUES A DATA TRANSFER REQUEST TO THE DMAC
S306
DMAC TRANSFERS PCM DATA TO THE INPUT FIFO
Yes
DMAC TRANSFER COMPLETION INTERRUPT IS ISSUED
S307
PCM DATA TRANSFER PROCESSING END

FIG.8


START OF CORRECTION OF DIFFERENCE OF THE ELAPSED TIMES BETWEEN THE BROADCAST STATION AND THE MOBILE TELEPHONE
THE INTERNAL REFERENCE CLOCK COUNT UNIT HOLDS THE COUNT VALUE OF THE CURRENT INTERNAL REFERENCE CLOCK BY USING THE DMAC TRANSFER COMPLETION INTERRUPT AS A TRIGGER
S400
THE COU CORE UNIT READS OUT THE COUNT VALUE OF THE INTERNAL REFERENCE CLOCK HELD AND CALCULATES PCR ADVANCE TIME
S401
STORE THE CALCULATED PCR ADVANCE TIME IN THE REFERENCE TIME STORAGE BUFFER
S402
THE CPU CORE UNIT CALCULATES THE AUDIO REPRODUCTION TIME
S403
THE CALCULATED AUDIO REPRODUCTION TIME IS ACCUMULATED IN THE AUDIO REPRODUCTION TIME ACQUIRED UP TO NOW AND STORED IN THE AUDIO REPRODUCTION TIME STORAGE BUFFER
S404
S405
PCR ADVANCE TIME = AUDIO REPRODUCTION TIME ?
No
Yes
S406
PCR ADVANCE TIME > AUDIO REPRODUCTION TIME ?
No
Yes
S407
CORRECT THE INPUT fs SET VALUE OF THE AUDIO CONTROL UNIT TO THE MINUS SIDE
S408
CORRECT THE INPUT fs SET VALUE OF THE AUDIO CONTROL UNIT TO THE PLUS SIDE
END OF CORRECTION OF THE DIFFERENCE OF ELAPSED TIMES BETWEEN THE BROADCAST STATION AND THE MOBILE TELEPHONE

BROADCAST STATION SYNCHRONIZATION METHOD AND MOBILE TERMINAL

INCORPORATION BY REFERENCE

The present application claims priority from Japanese application JP2004-315589 filed on Oct. 29, 2004, the content of which is hereby incorporated by reference into this application.

BACKGROUND OF THE INVENTION

The present invention relates to a broadcast station synchronization technique in a digital TV broadcast reception system and in particular, to a broadcast station synchronization method for comparing the reference time information of the broadcast station side to the audio reproduction time of the mobile terminal which receives the information, acquiring a difference of the time elapse of them, and performing control to adjust the audio reproduction time, thereby eliminating the difference and a technique which can effectively be applied to the mobile terminal.

The inventor of the present invention has examined the following techniques associated with the broadcast station synchronization in the conventional digital TV broadcast reception system.

For example, as a synchronization method for a digital TV broadcast receiver with a broadcast station, JP-A-2004-23136 (FIG. 1) suggests a method for using a inexpensive non-feedback type oscillator, instead of a voltage feedback type oscillator such as a VCO (Voltage Controlled Oscillator), to compare the broadcast station reference time information PCR (Program Clock Reference) to a reproduction audio time stamp so as to detect a difference and correct the difference, thereby synchronizing with the broadcast station.

SUMMARY OF THE INVENTION

However, the inventor of the present invention has found that the aforementioned broadcast station synchronization technique in the digital TV broadcast reception system has problems as follows.

For example, since the system clock of the mobile phone has a frequency mainly for communication, it is not synchronized with and has a different frequency from the broadcast station reference clock of 27 MHz required for receiving digital TV broadcast. By mounting a VCO and a PLL of 27 MHz like a fixed receiver, it is easy to synchronize with the broadcast station but the cost and the mobile telephone size are increased and there arises a problem such as an unnecessary radiation caused by addition of another clock. Furthermore, additional parts used increase the power consumption and the viewing-enabled time is shortened.

Moreover, in the broadcast station synchronization method disclosed in JP-A-2004-23136, only one clock is used to synchronize with the broadcast station and the number of data of the PCM data is adjusted so that audio reproduction is performed with sampling frequency fs inherent to the PCM data. Accordingly, adjustment of the number of data causes overlapped reproduction of the same data or thinning of data in the middle when reproduced. That is, distortion is caused in reproduced audio.

It is therefor an object of the present invention to provide a broadcast station synchronization technique capable of improving the synchronization accuracy of the broadcast station synchronization while suppressing the distortion of the reproduced audio by using only the communication system clock provided in the mobile terminal without adding parts such as a 27-MHz VCO to the mobile terminal for receiving a digital TV broadcast in the mobile terminal.

The aforementioned object and the other objects of the present invention will be made clear by the description of the present Specification and the attached drawings.

The inventions disclosed in this application can be outlined as follows.

According to the present invention, the fs setting value of the PCM data to be reproduced is adjusted so as to eliminate a temporal difference between the audio reproduction at the mobile terminal and the reference clock of the broadcast station, thereby performing synchronization. Here, for performing a comparison to decide whether a difference is caused, the aforementioned two time information for a certain period are obtained more accurately, thereby achieving the aforementioned object.

For this, a DMAC transfer completion interrupt of the DMAC (Direct Memory Access Controller) used for PCM data transfer is used as a latch trigger of the internal reference clock count unit reproducing the clock synchronized with the broadcast station. Thus, it becomes possible to obtain the reference clock time of the broadcast station identical to the audio reproduction time and acquire the aforementioned two times within the same period more accurately.

Moreover, as a method for correcting the difference between the aforementioned two times, instead of increasing or reducing the number of PCM data outputted to the audio-DAC, the fs value in the audio reproduction unit is varied so as to adjust the PCM data timing outputted to the audio-DAC.

In this case, the fs value for outputting the PCM data to the audio-DAC is adjusted in plus or minus direction within a range permitted for human auditory sense characteristics for the original fs, so that the output time can be made later/earlier as compared to the time when the output is performed to the audio-DAC with the fs inherent to the PCM data. Accordingly, without increasing or decreasing the number of PCM data, it is possible to adjust the audio reproduction time and to perform audio reproduction while suppressing the distortion. Furthermore, there is no uncomfortable feeling attributed to the correction of the fs when performing audio-DAC transfer.

By using the aforementioned method, it is possible to perform synchronization with a broadcast station while suppressing the distortion of the reproduced audio and provide a small-size mobile terminal at a reasonable cost.

BRIEF DESCRIPTION OF THE DRAWINGS

There and other features, objects and advantages of the present invention will become more apparent from the following description when taken in conjunction with the accompanying drawings wherein:

FIG. **1** is a block diagram showing an example of configuration of a mobile telephone according to an embodiment of the present invention.

FIG. **2** shows an example of configuration of an application CPU required for receiving a digital TV broadcast and an example of video/audio data flow in the embodiment of the present invention.

FIG. **3** is a block diagram showing an example of configuration of a TS processing unit in the embodiment of the present invention.

FIG. **4** is a block diagram showing an example of configuration of the audio control unit in the embodiment of the present invention.

FIG. **5** is a main flowchart showing an example of series of operations from the view start in the embodiment of the present invention.

FIG. **6** is a flowchart showing the operation of the mobile telephone internal reference clock to follow the PCR of the broadcast station processed in the TS processing unit in the broadcast synchronization processing in the embodiment of the present invention.

FIG. **7** is a flowchart showing a detailed example of the operation of "the PCM data transfer processing" in the main flowchart of FIG. **5** in the embodiment of the present invention.

FIG. **8** is a flowchart showing a detailed example of the operation of "the processing for correcting the difference between the time elapses of the broadcast station and the mobile telephone."

DESCRIPTION OF THE EMBODIMENTS

Description will now be directed to embodiments of the present invention with reference to the attached drawings. It should be noted that in principle, like members are denoted by like reference symbols and their repeated explanations are omitted. The present invention can be applied to mobile terminals in general such as a mobile telephone, a PHS (Personal Handy-phone System), and a PDA (Personal Digital Assistant) but explanation will be given on the mobile telephone as an example.

As an embodiment of the present invention, referring to FIG. **1** to FIG. **8**, explanation will be given on the method for improving synchronization accuracy in the broadcast station synchronization when receiving a digital TV broadcast by using a mobile terminal not having a clock generator of frequency identical to the broadcast station reference clock of 27 MHz.

The present embodiment is applied to a mobile terminal having a digital TV broadcast reception function. Here, explanation will be given on a case using a mobile telephone as an example of the mobile terminal. However, this does not limit the device of the reception side applied to the present invention. The same configuration can be used for other than the mobile telephone.

FIG. **1** is a block diagram showing an example of configuration of a mobile telephone according to an embodiment of the present invention.

The mobile telephone in the present embodiment includes an antenna **1**, a radio communication unit **2**, an encoding/decoding unit **3**, a microphone **4**, a receiver **5**, a communication CPU **6**, a communication memory **7**, an input unit **8**, a system clock **9**, an application CPU **10**, an application CPU memory **11**, a work memory **12**, a display device **13**, an audio-DAC (Digital Analog Converter) **14**, a loudspeaker **15**, a digital TV broadcast reception antenna **20**, a digital TV broadcast radio-frequency unit **21**, and the like.

The antenna **1** converts an electric wave into a high-frequency electric signal and performs opposite conversion. The radio communication unit **2** demodulates the radio-frequency electric signal and modulates the output signal of the encoding/decoding unit **3**. The encoding/decoding unit **3** decodes the output signal of the radio communication unit **2**, outputs it to the receiver **5** and the communication CPU **6**, encodes the output signal of the communication CPU **6** and the microphone **4**, and outputs it to the radio communication unit **2**. The communication CPU **6** mainly performs processing control associated with the communication of the telephone. The communication memory **7** stores a program of the communication CPU **6** and the like. The input unit **8** may include a numeric key and a cursor key. The system clock **9** is a clock signal of the mobile telephone which is different from the 27-MHz clock.

The application CPU **10** mainly performs processing and control associated with the application of the mobile telephone. The application CPU memory **11** stores a program of the application CPU and the like. The work memory **12** is a memory used as a work area when executing an application. The display device **13** is an LCD (Liquid Crystal Display) or the like. The audio-DAC **14** converts PCM data into an analog audio signal.

The digital TV broadcast reception antenna **20** receives an electric wave of the digital TV broadcast and converts it into a radio-frequency electric signal. The digital TV broadcast radio-frequency unit **21** includes a radio-frequency circuit for demodulating a radio-frequency electric signal outputted from the digital TV broadcast reception antenna **20**.

FIG. **2** shows an example of configuration of the application CPU **10** required for receiving a digital TV broadcast and an example of video/audio data flow.

The application CPU **10** includes: a TS processing unit **200** for isolating a broadcast station reference time information PCR (Program Clock Reference) from the MPEG2-transport stream (hereinafter, referred to as MPEG-TS) which is a protocol used in the digital broadcast specification outputted from the digital TV broadcast high-frequency unit **21**; a TS-DeMux unit **201** for separating the MPEG2-TS into a moving picture elementary stream (hereinafter, referred to as video-ES) and audio elementary stream (hereinafter, referred to as audio-ES); a video decoding unit **202** for decoding the separated video-ES and converting it into plot data which can be displayed on the display device **13**; a display control unit **203** for transmitting the plot data to the display device **13**; an audio decoding unit **204** for decoding the separated audio-ES and converting it into PCM data; an audio control unit **205** for transmitting the PCM data to the audio-DAC **14** and setting the sampling frequency (hereinafter, referred to as fs) when reproducing the PCM data; a CPU core unit **206** for controlling the aforementioned processes and calculating the audio reproduction time and the like; and a DMAC **207** for transmitting the PCM data at a high speed instead of the CPU between the work memory **12** and the audio control unit **205**, for example. The application CPU **10** further includes a TS buffer **300** for temporarily holding various data generated by the aforementioned processes, a video-ES buffer **301**, a plot data buffer **302**, an audio-ES buffer **303**, and PCM data buffer **304**.

The TS processing unit **200** is supplied with the system clock **9** as a source clock for generating the internal reference clock and a DMAC transfer completion interrupt outputted from the DMAC **207** upon completion of the PCM data transfer of the set number of samples. Moreover, at an arbitrary timing, it is possible to read out a value from the CPU core unit **206**.

The DMAC **207** can set the number of DMA (Direct Memory Access) transfers and transfers the PCM data of the number of samples for the set number of times. The DMA transfer is performed repeatedly until the digital TV broadcast ends or the user performs the viewing end operation. Moreover, when the PCM data transfer of the set number of samples is complete, the DMAC **207** outputs a DMAC transfer completion interrupt. The DMAC transfer completion interrupt is inputted to the CPU core unit **206** so as to be used as a timing signal calculating the audio reproduction time by the CPU core unit **206**. Furthermore, the DMAC transfer completion interrupt is also inputted to the TS (Transport Stream) processing unit **200** and used for latching the count value of the internal reference clock count unit **402** which will be detailed later. Thus, it is possible to acquire the elapse time of the clock synchronized with the broadcast station at the same moment as the audio reproduction time and obtain the two times within the same period more accurately, thereby improving the synchronization accuracy between the broadcast station and the mobile telephone.

Each of the buffers **300**, **301**, **302**, **303**, and **304** handles large-size data and in general, the work memory **12** is divided into several parts to be allocated for various buffers. However, it is also possible to use the built-in memory provided in the application CPU. Moreover, it is possible to allocate each of the buffers by dividing them into the external work memory **12** and the internal memory.

FIG. **3** shows an example of configuration of the TS processing unit **200**.

The TS processing unit **200** includes: a PCR (Program Clock Reference) extraction unit **400** for isolating a PCR from the MPEG2-TS inputted from the digital TV broadcast high frequency unit **21**; an internal reference clock generation unit **401** for generating an internal reference clock from the system clock of the mobile telephone; an internal reference clock count unit **402** for reproducing the clock synchronized with the broadcast station by the correction by the PCR; a comparison unit **403** for comparing the PCR value outputted from the PCR extraction unit **400** to thee count value of the internal reference clock count unit **402** and outputting count value correction value information to the internal reference clock count unit **4402** when the two values are different from each other; and an internal reference clock count value holding unit **404** for holding the count value latched by the internal reference clock count unit **402** by using the DMAC transfer completion interrupt signal as a trigger and enabling read out from the CPU core unit **206** at an arbitrary timing.

The internal clock count unit **402** counts up by the internal reference clock and when the count value is different from the PCR value, the count value is replaced by the PCR value according to the count value correction information. Since the PCR is broadcast station reference clock information, if the internal reference clock count unit **402** can perform count up operation to follow the PCR, it means that the internal reference clock count unit **402** reproduces the time elapse following the clock synchronized with the broadcast station.

Since the correction of the internal reference clock count unit **402** is performed only when the PCR is received, no correction is performed while no PCR is received. During the period when no correction by the PCR is performed, it is assumed that the internal reference clock advances like the clock synchronized with the broadcast station, and only the count up by the internal reference clock is performed.

FIG. **4** shows an example of configuration of the audio control unit **205**.

The audio control unit **205** includes a PCM data processing unit **500** for setting fs of the PCM data, stereo/monaural, etc; a DAC-IF unit **501** for converting the PCM data into LRCLK/BCCLK/Data so as to be outputted to the audio-DAC **14**; an input FIFO (First In First Out) **600** for buffering the input/output PCM data; and an output FIFO **601**.

The fs should be set in advance for the PCM data processing unit **500**. In order to perform accurate audio reproduction, it is necessary to accurately set the fs of the PCM data. However, there is no need of setting the same value as the fs of the PCM data. Details will be given later. For example, the relationship between the input fs and the audio reproduction time is as follows. For the input fs=24 k [Hz] (=audio reproduction time without correction of the fs), the following adjustment is made: input fs=24 k+α [Hz] (=fs is corrected to shorten the audio reproduction time), input fs=224 k−α [Hz] (=fs is corrected to increase the audio reproduction time).

Moreover, the PCM data processing unit **500** can performs fs conversion by the filtering processing or the like so that output is enabled with an output fs different from the fs of the inputted PCM data to the audio-DAC. This is a method effective for a mobile telephone having no master clock for audio reproduction. Like the broadcast station synchronization, the fs which can be generated by the system clock of the mobile telephone is set as an output fs in the PCM data processing unit **500** for performing the PCM data reproduction.

Since the input FIFO **600** has a very small capacity as compared to the length of the content to be reproduced, it is necessary to monitor the data in the input FIFO **600** so that no underflow or overflow is caused and if necessary, transfer of the next data is requested to the DMAC **207**.

Next, referring to FIG. **5** to FIG. **8**, explanation will be given on a method for improving the accuracy of the synchronization with the broadcast station by using operation flow for synchronization with the broadcast station required for receiving the digital TV broadcast by a mobile telephone.

FIG. **5** is a main flowchart indicating an example of series of operations from the view start. FIG. **6** is a flowchart showing an example of operation for making the internal reference clock of the mobile telephone follow the PCR of the broadcast station processed in the TS processing unit **200** in the broadcast station synchronization processing. FIG. **7** is a flowchart showing details of the operation of "the PCM data transfer processing" in the main flowchart of FIG. **5**. FIG. **8** is a flowchart showing details of the operation of "the processing for correcting the difference of the time elapse between the broadcast station and the mobile telephone" in the main flowchart of FIG. **5**.

Firstly, as shown in FIG. **5**, when the user operates the input unit **8** to start viewing of the digital TV broadcast, channel information is interpreted from the electric wave received by the digital TV broadcast antenna **20** and initialization is performed (S**100**). Subsequently, the reference time storage buffer and the audio reproduction time storage buffer required for performing the broadcast station synchronization are initialized (S**101**). These two buffers are normally allocated in the work memory **12** but if the memory built in the application CPU **10** is available, it can be used.

The TS processing unit **200** extracts MPEG2-TS from the received electric wave and stores it in the TS buffer **300** (S**102**). Here, the PCR required for the internal reference clock count unit **402** to follow the clock synchronized with the broadcast station is also extracted. The MPEG2-TS stored in the TS buffer **300** is divided into the video-ES and the audio-ES by the TS-DeMux unit **201** so as to be stored in the video-ES buffer **301** and the audio-ES buffer **303**, respectively (S**103**). Since the AV synchronization is based on the audio reproduction timing, while audio processing is performed, the video-ES stored in the video-ES buffer is decoded by the decoding unit **202** and transferred from the display control unit **203** to the display device **13** in accordance with the audio reproduction timing, thereby performing plot processing (S**108**).

The audio-ES stored in the audio-ES buffer **303** is decoded by the audio decoding unit **204**, converted into the PCM data, and stored in the PCM data buffer **304** (S**104**). The PCM data in the PCM data buffer **304** is transferred to the audio control unit **205** by the DMAC **207** (S**105**) and audio reproduction is performed (S**107**). Here, the audio reproduction time is compared to the count up time of the internal reference clock count unit **4402** reproducing the clock synchronized with the broadcast station so as to obtain a difference (delay/advance) of an audio reproduction time. The input fs set in the PCM data processing unit **500** is corrected so as to eliminate the difference (S**106**)and the audio reproduction time of the mobile telephone is telephone is matched with the time elapse of the broadcast station, thereby synchronizing the mobile telephone with the broadcast station for viewing the program. Here, the correction width of the input fs is set within a range permissible for human auditory characteristic, thereby suppressing the reproduced audio distortion or artificiality. The S**105** and S**106** are the point for improving the synchronization accuracy of the broadcast station synchronization and they will be detailed with reference to FIG. **6** and FIG. **7**.

When terminating the viewing of the digital TV broadcast, the user presses the end key (S**109**). If the end key is not pressed (No in S**109**), the MPEG2-TS is again extracted from the broadcast signal wave (S**102**) and viewing is continued while maintaining synchronization with the broadcast station. When the end key is pressed (Yes in S**109**), all the processes are terminated and the viewing is terminated.

Subsequently, referring to FIG. **6**, explanation will be given on the operation for making the internal reference clock of the mobile telephone follow the PCR of the broadcast station.

As shown in FIG. **6**, when the PCR following operation is started, the internal reference clock is generated using the system clock **9** inputted to the internal reference clock generation unit **401** as a source clock. The internal reference clock count unit **402** performs count up operation by the internal reference clock (S**200**). The PCR extraction unit **400** acquires the PCR superimposed on the MPEG2-TS (S**201**). This is used to make the internal reference clock follow the clock synchronized with the broadcast station. When the PCR cannot be extracted, the internal reference clock count unit **402** does not perform correction processing and continues only the count up operation by the internal reference clock (No in S**201**).

When the PCR can be extracted (Yes in S**201**), the value of the extracted PCR is compared to the count value of the current internal reference clock count unit **402** (S**202**). If the two values coincide (Yes in S**202**), it is judged that the time of the clock synchronized with the broadcast station and the time of the internal reference clock advance in the same way and the count up operation by the internal reference clock is continued until the next PCR is received without correcting the count value of the internal reference clock count unit **402**.

When the two values do not coincide (No in S**202**), it is judged that there is a difference between the time of the clock synchronized with the broadcast station and the time of the internal reference clock and there arises a need of matching the internal reference clock with the clock synchronized with the broadcast station. For this, the count value of the current internal reference clock count unit **4402** is replaced by the value of the PCR extracted by the PCR extraction unit **400** (S**203**) and the count up operation by the internal reference clock is continued until the next PCR is received. The aforementioned PCR following processing of the internal reference clock continues without halt while receiving the digital TV broadcast (No in S**204**). When the end key is pressed (Yes in S**204**), the PCR following operation terminates.

Subsequently, referring to FIG. **7**, explanation will be given on the PCM data transfer processing flow (S**105**) in the main flowchart of FIG. **5**.

As shown in FIG. **7**, when the PCM data transfer processing is started, the PCM data decoded by the audio decoding unit **204** is stored in the PCM data buffer **304** and during the decoding, the parameter extractions from the audio data is performed. The CPU core unit **206** sets the parameters in the audio control unit **205** (S**300**). The parameters set here include the input fs value and stereo/monaural information. Moreover, the CPU core unit **206** also sets the number of transferred samples and the transfer method in the DMAC **207**.

When the parameter setting is complete and the PCM data is stored in the PCM data buffer **304**, the CPU core unit **2206** instructs the DMAC **207** to start the PCM data transfer (S**301**). Here, the CPU core unit **206** stores the count value of the internal reference clock count unit **402** at the PCM data transfer start (S**302**). The count value at the PCM data transfer start is used when calculating the time elapse of the clock synchronized with the broadcast station.

When the DMAC receives the PCM data transfer start instruction from the CPU core unit **206**, it successively transfers the PCM data stored in the PCM data buffer **304** to the input FIFO **600** of the audio control unit **205** (S**303**). The PCM data transferred to the input FIFO **600** is transferred to the audio-DAC **14** at the cycle of the input fs value set in S**300**. After this, it is checked whether the PCM data corresponding to the number of samples set in S**300** has been transferred (S**304**).

When transfer of the set number of samples has not been completed (No in S**304**), audio reproduction is performed so as to reduce the PCM data in the input FIFO **600**. The audio control unit **205** issues the next PCM data transfer request to the DMAC **207** (S**305**) and the next PCM data is transferred from the PCM data buffer **304** to the input FIFO **600** (S**306**). The steps S**304** and S**305** are repeatedly performed until the number of samples set in S**300** are transferred to the audio control unit **205**.

When the transfer of the set number of samples is complete (Yes in S**304**), the DMAC **207** outputs a DMAC transfer completion interrupt to the CPU core unit **206** and the internal reference clock count unit **402** (S**307**). Control is returned to the main flow of FIG. **5** so as to perform the synchronization processing with the broadcast station as the next step (S**106**).

After output of the DMAC transfer completion interrupt, control is returned temporarily to the main flow for performing the synchronization processing with the broadcast station (S**106**). However, while viewing of the digital TV broadcast continues, control is immediately returned to the PCM data transfer processing flow and the next PCM data transfer processing is performed before the audio to be reproduced is terminated.

In the PCM data transfer processing performed for the second time and after, the input fs set value for adjusting the audio reproduction time has been corrected. By correcting the input fs set value, there arises a difference between the time when the PCM data is reproduced and the time when actually outputted to the audio-DAC **14** to be reproduced. This difference is used to correct the difference between the time elapse of the clock synchronized with the broadcast station reproduced by the internal reference clock count unit **402** and the audio reproduction time of the mobile telephone, thereby performing synchronization with the broadcast station.

Subsequently, referring to FIG. **8**, explanation will be given on the processing flow (S**106**) for correcting the difference between the time elapse of the broadcast station and that of the mobile telephone in the main flow of FIG. **5**.

When this processing flow is executed, the internal reference clock count unit **402** follows the clock synchronized with the broadcast station by the PCR following operation of the internal reference clock shown in FIG. **6**. Furthermore, by the PCM data transfer processing shown in FIG. **7**, the PCM data of the number of the samples set in S**300** is transferred to the audio control unit **205** and the DMAC transfer completion interrupt is outputted from the DMAC **207** to the CPU core unit **206** and the internal reference clock count unit **402**.

Only the internal reference clock count unit **402** reproduces the clock synchronized with the broadcast station in the mobile telephone. In order to increase the synchronization accuracy, it is necessary to acquire more accurate audio reproduction time and the elapsed time of the clock synchronized with the broadcast station. Within the same period, the time required for reproducing the predetermined number of samples of the PCM data in the mobile telephone is used as the period for acquiring the more accurate audio reproduction time and the elapsed time of the clock synchronized with the broadcast station. Since the time required for reproducing the PCM data is the most accurate and stable time in the mobile telephone, the time required for reproducing a predetermined number of samples of PCM data is used as a unit period for comparing the elapsed time of the clock synchronized with the broadcast station to the audio reproduction time of the mobile telephone.

As for the method for accurately acquiring the elapsed time of the clock synchronized with the broadcast station and the audio reproduction time within the same period, the audio reproduction time of the mobile telephone can be calculated by Expression (1) according to the fs set value of the PCM data to be reproduced and the number of samples as follows.

$$\text{audio reproduction time} = 1/\text{fs set value} \times \text{number of samples [sec]} \quad (1)$$

On the other hand, the time per unit period in the elapsed time of the clock synchronized with the broadcast station is made a reference time, which can be calculated by following Expression (2) according to the count value at the transfer start point of the PCM data in the internal reference clock count unit **402** and the count value at the moment of completion of transfer of a predetermined number of samples of PCM data to the audio control unit **205** as follows.

$$\text{reference time} = (\text{count value at transfer start} - \text{count value at transfer end}) \times 1/\text{internal reference clock [sec]} \quad (2)$$

For this, it is necessary to obtain the count value of the internal reference clock count unit **402** at the moment of completion of transfer of the predetermined number of samples of PCM data. When this count value is near to the moment of completion of transfer of the PCM data, it is possible to improve the accuracy of the elapsed time of the clock synchronized with the broadcast station. For this, the PCM data transfer completion interrupt signal is inputted directly to the internal reference clock count unit **402** so as to latch the internal reference clock count unit **402** by the interrupt signal at the moment of transfer completion, so that the count value is held by the internal reference clock count value holding unit **404**. Thus, it becomes possible to obtain the count value of the internal reference clock count unit **402** at the moment of transfer completion, thereby improving the synchronization accuracy.

As shown in FIG. **8**, when correction of the difference in the elapsed time between the broadcast station and the mobile telephone is started, the internal reference clock count unit **402** latches the count value by using the DMAC transfer completion interrupt as a trigger and holds it in the internal reference clock count value holding unit **404** (S**400**). The CPU core unit **206** reads out the count value upon generation of the DMAC transfer completion interrupt held in the internal reference clock count value holding unit **404** and calculates the reference time by using Expression (2) (S**401**).

The reference time is the a time from the PCM data transfer start to the output of the DMAC transfer completion interrupt signal and can be obtained by a difference between the count value of the internal reference clock latched by using the DMAC transfer completion interrupt signal as a trigger and the count value of the internal reference clock immediately after the start of the PCM data transfer. The calculated reference time is stored in a reference time storage buffer (S**402**).

Subsequently, the CPU core unit **206** calculates the audio reproduction time by using Expression (1) (S**403**). The reference time is a time from the PCM data reproduction start to the current time while the audio reproduction time is a time for a predetermined number of samples and not the time from the reproduction start. The CPU core unit **206** accumulates the audio reproduction time for a predetermined number of samples obtained by the calculation from the PCM data transfer start and stores it in an audio reproduction time storage buffer (S**404**).

Next, check is made to detect whether a difference is present between the reference time obtained from the clock synchronized with the reproduced broadcast station and the audio reproduction time of the mobile telephone (S**405**). When no difference is present (Yes in S**405**), it is judged that the time advance in the broadcast station is identical to the time advance of the mobile telephone. Control is returned to the main flow of FIG. **5** without performing any correction and the audio reproduction operation is continued.

When a difference is present (No in S**405**), the input fs set value of the PCM data processing unit **500** upon reproduction of the PCM data is corrected so that the audio reproduction time advance of the mobile telephone is matched with the time advance of the broadcast station, thereby adjusting the audio reproduction time. Firstly, in order to correct the input fs set value of the PCM data processing unit **500**, the advance of the audio reproduction time with respect to the reference time is checked. According to the check result, it is judged whether the input fs set value is corrected to the plus side or the minus side (S**406**).

When the audio reproduction time advances forward (Yes in S**406**), the PCM data reproduction time is faster and accordingly, the input fs set value at the next PCM data transfer is corrected to the minus side than the current input fs set value (S**407**). On the contrary, when the reference time advances forward (No in S**406**), the PCM data reproduction time is slower and accordingly, the input fs set value at the next PCM data transfer is corrected to the plus side than the current input fs set value (S**408**). The fs correction width here is assumed to be within a range permissible for the human auditory characteristics. 24 kHz and the number of samples transferred by the DMAC is **1024** samples, the PCM data of fs=24 kHz is reproduced at the timing of the input fs=24 kHz when no correction is performed by the input fs, and the PCM data reproduction time of the 1-24 samples becomes $1/24000 \times 1024 \approx 42.667$ msec.

As compared to this, when the audio reproduction time advances forward, the reproduction time of the PCM data is faster and the input fs set value is corrected to the minus side so as to delay the PCM data reproduction time. For example, when 10 Hz correction is performed to the minus side, the PCM data of fs=24 kHz is reproduced at the timing of the input fs=23.990 kHz and the reproduction time of the 1024-sample PCM data becomes $1/23990 \times 10224 \approx 42.684$ msec. As compared to the case when no correction of the input fs is performed, the reproduction time can be delayed by about 0.17 msec.

On the contrary, when the reference time advances forward, the PCM data reproduction time is slower and the input fs set value is corrected to the plus side so that the PCM data reproduction time advances forward. For example, when 10 Hz correction is performed to the plus side, the PCM data of fs=24 kHz is reproduced at the timing of fs=24.010 kHz and the 1024-sample PCM data reproduction time becomes $1/24010 \times 1024 \approx 42.649$ msec. As compared to the case when no correction of the input fs is performed, the when no correction of the input fs is performed, the reproduction time can advance forward by about 0.18 msec.

By adjusting the audio reproduction time by the aforementioned method, it is possible not to change the number of data in the PCM data actually reproduced. Furthermore, the fs correction width can be set within a predetermined range permissible for the human auditory characteristic. Thus, it is possible to suppress the distortion and artificiality of the reproduced audio. In general, the range permissible for the human auditory characteristic is in the order of 0.2%. For example, when fs=24 kHz, the fs correction width is about 24 kHz±48 Hz.

According to the present embodiment, it becomes possible to realize improvement of synchronization accuracy in synchronization with the broadcast station in the mobile terminal while suppressing the distortion of the reproduced audio without using an additional part such as a VCO of 27 MHz. Thus, no additional part is used in the mobile terminal, it is possible to provide a small-size mobile terminal at a reasonable cost.

Furthermore, since increase of additional current consumption can be suppressed, it is possible to view a digital TV broadcast for a long continuous time.

Moreover, by performing the fs correction auditory characteristic, it is possible to suppress the affect of the pitch difference with respect to the reproduced audio by the correction.

The invention made by the inventors and thus far explained through the embodiment is not limited to the embodiment and can be modified without departing from the spirit of the invention.

The present invention relates to a broadcast station synchronization technique in the digital TV broadcast reception system, for example, and in particular to a mobile terminal having the digital TV broadcast reception function, and can be applied to a mobile telephone and reception side device having the similar configuration in general such as PHS and PDA.

While we have shown and described several embodiments in accordance with our invention, it should be understood that disclosed embodiments are susceptible of changes and modifications without departing from the scope of invention. Therefore, we do not intent to be bound by the details shown and described herein but intend to cover all such changes and modifications within the ambit of the appended claims.

The invention claimed is:

1. A broadcast station synchronization method in a mobile terminal which is mobile relative to a broadcast station and operating with a system clock different from a reference clock of the broadcast station, in which a received signal is decoded to produce a PCM data and the PCM data is transferred to a digital/analog converter for reproducing an audio signal,

wherein a clock synchronized with the broadcast station is reproduced according to an internal reference clock generated by the system clock of the mobile terminal and corrected by a reference time information from the broadcast station,

an elapsed time is compared to audio signal reproduction time at a predetermined time interval so as to detect a delay or advance difference of the audio signal reproduction time with respect to the elapsed time,

the audio signal reproduction time is adjusted so as to cancel the difference, and

in the transferring of the PCM data to the digital/analog converter for reproducing the audio signal, a signal indicating a completion of the transferring of the PCM data of a predetermined number of samples is used as a trigger for latching an internal reference clock count unit reproducing the clock synchronized with the broadcast station so as to acquire the clock time synchronized with the broadcast station of the same period as the audio signal reproduction time per a predetermined period, thereby performing synchronization of the reproduced audio signal with the broadcast station.

2. A broadcast station synchronization method in a mobile terminal which is mobile relative to a broadcast station and operating with a system clock different from a reference clock of the broadcast station, in which a received signal is decoded to produce a PCM data and the PCM data is transferred to a digital/analog converter (DAC) for reproducing an audio signal,

wherein a clock synchronized with the broadcast station is reproduced according to an internal reference clock generated by the system clock of the mobile terminal and corrected based on a reference time information from the broadcast station,

an elapsed time is compared to audio signal reproduction time at a predetermined time interval so as to detect a delay or advance difference of the audio signal reproduction time with respect to the elapsed time,

the audio signal reproduction time is adjusted so as to cancel the difference, and

during audio signal reproduction, a number of data in PCM data to be reproduced is not changed and the sampling frequency when outputting the PCM data to the DAC is adjusted, thereby performing synchronization of the reproduced audio signal with the broadcast station.

3. A broadcast station synchronization method in a mobile terminal which is mobile relative to a broadcast station and operating with a system clock different from a reference clock of the broadcast station, in which a received signal is decoded to produce a PCM data and the PCM data is transferred to a digital/analog converter (DAC) for reproducing an audio signal,

wherein a clock synchronized with the broadcast station is reproduced according to an internal reference clock generated by the system clock of the mobile terminal and corrected by a reference time information from the broadcast station,

an elapsed time is compared to audio signal reproduction time at a predetermined time interval so as to detect a delay or advance difference of the audio signal reproduction time with respect to the elapsed time,

the audio signal reproduction time is adjusted so as to cancel the difference,

in the transferring of the PCM data to the digital/analog converter for reproducing the audio signal, a signal indicating a completion of the transferring of the PCM data of a predetermined number of samples is used as a trigger for latching an internal reference clock count unit reproducing the clock synchronized with the broadcast station so as to acquire the clock time synchronized with the broadcast station of the same period as the audio signal reproduction time per a predetermined period, and

during audio signal reproduction, a number of data in the PCM data to be reproduced is not changed and the sampling frequency when transferring the PCM data to the DAC is adjusted, thereby performing synchronization of the reproduced audio signal with the broadcast station.

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